

# VoipSwitch(Soft Switch) + SiSky (Skype Landline Gateway) Solution

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**Yeastar Technology Co., Ltd.**

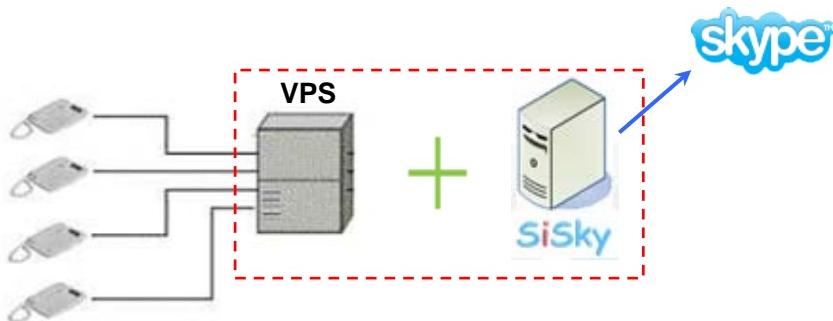
# 1. Introduction

## 1.1 Solution Overview

SiSky is a 100% software Skype gateway that supports maximum 30 channels on a single computer. Utilizing SiSky gateway with soft switch, businesses can substantially save on both domestic long-distance and international calls.

Firstly install SiSky on a server and then dock the soft switch in the IP address of SiSky server, users can make the calls to any landline number (PSTN & Mobile) at low SkypeOut rates on the normal desk phone without any habits changed.

## 1.2 Flow Chart



## 1.3 Environment

### **SiSky Server Requirements:**

CPU	Intel Xeon X5410 * 2
Mainboard	MCH Intel 5000V ICH Intel 6321ESB
Memory	ECC 2G * 2
Disk	SATA 80G 7200rpm
Power Supply	AC Input: 100~240V, 50~60Hz,3~6A; AC Output: 600W (MAX)
Network	RJ45 (100M Ethernet)
OS	Windows XP / 2003
Internet Connection	Dedicated Broadband downlink >=2Mbps, uplink >=1Mbps

**Soft Switch Platform:**

Type	VoipSwitch
------	------------

***Best Application***

Skype Unlimited Calls Subscriptions

***Management***

Load Balancing: system will distribute the call rate equally among all trunks

Correct Billing System: Whether to not to answer SIP incoming calls immediately. System takes with reversal signal that can charge the calling time correctly.

***Compatibility***

Compatible with all soft switch platforms support with standard SIP protocol

***Carrier Class Stabile Performance***

30 Channels in a single server is unique to SiSky Skype gateway.

Non-destructive control of Skype voice stream to assure the excellent call quality.

## 2. Installing SiSky Server

### 2.1 Installing SiSky Software

This section shows how to install SiSky EE software on the PC.

Note: Before installing SiSky software, you should first **uninstall** Skype software on your computer if you already have Skype software installed.

**Steps:**

1. Download software from website <http://www.yeastar.com>. And Double-Click to Start Installation Process.
2. A **Welcome to the SiSky Installation** screen will come up. Click **Next** to continue. See Figure 2-1.

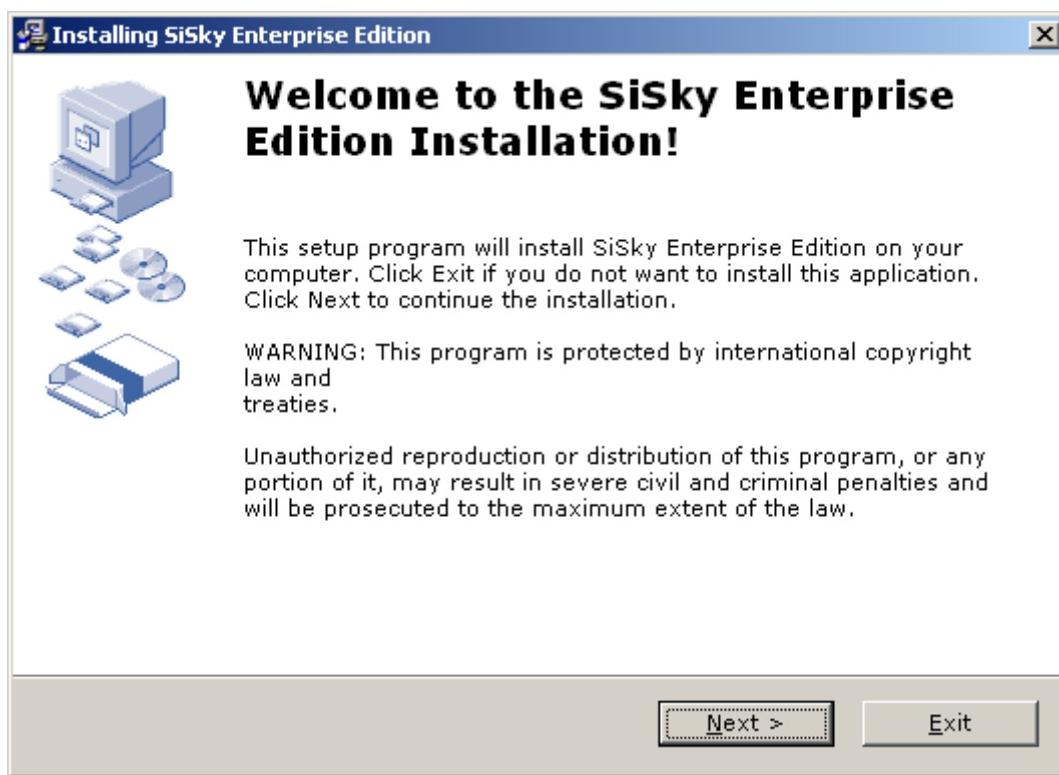


Figure 2-1

3. Read through the software **License Agreement**, select **I agree with the above terms and conditions**, and then click **Next** to continue. See Figure 2-2

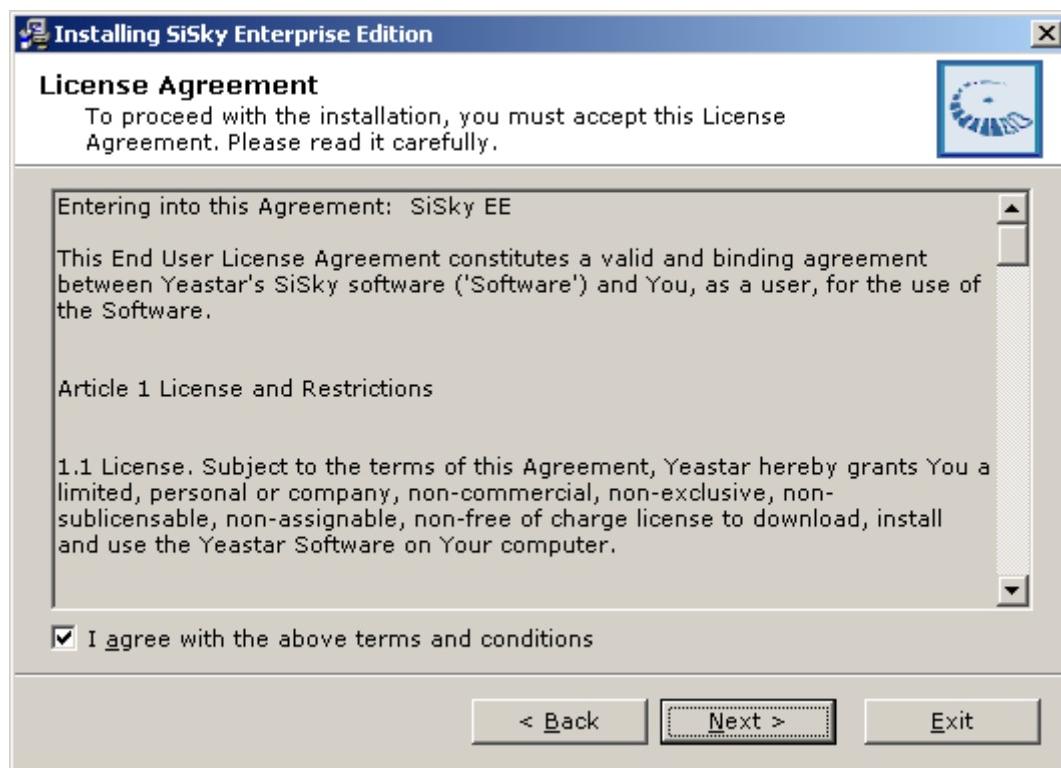


Figure 2-2

4. A Readme info window will appear. Click **Next** to continue.

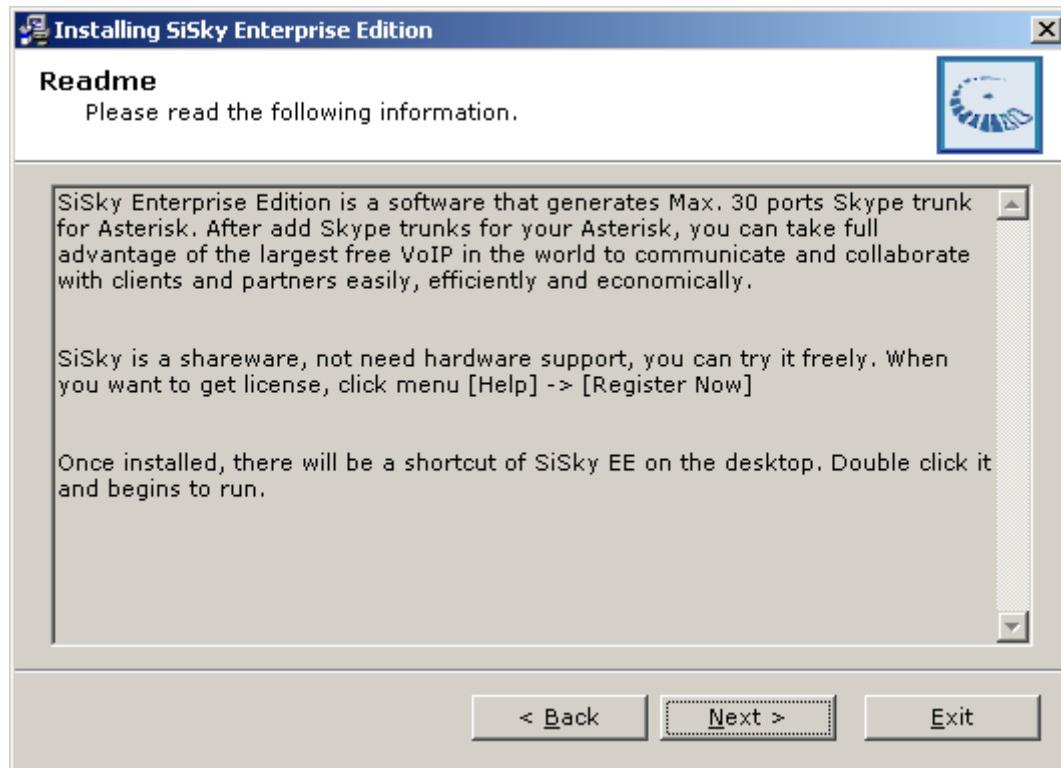


Figure 2-3

5. The **Destination folder** window will offer you the option where you would like SiSky to

be stored on your computer. Click **Next** to continue.

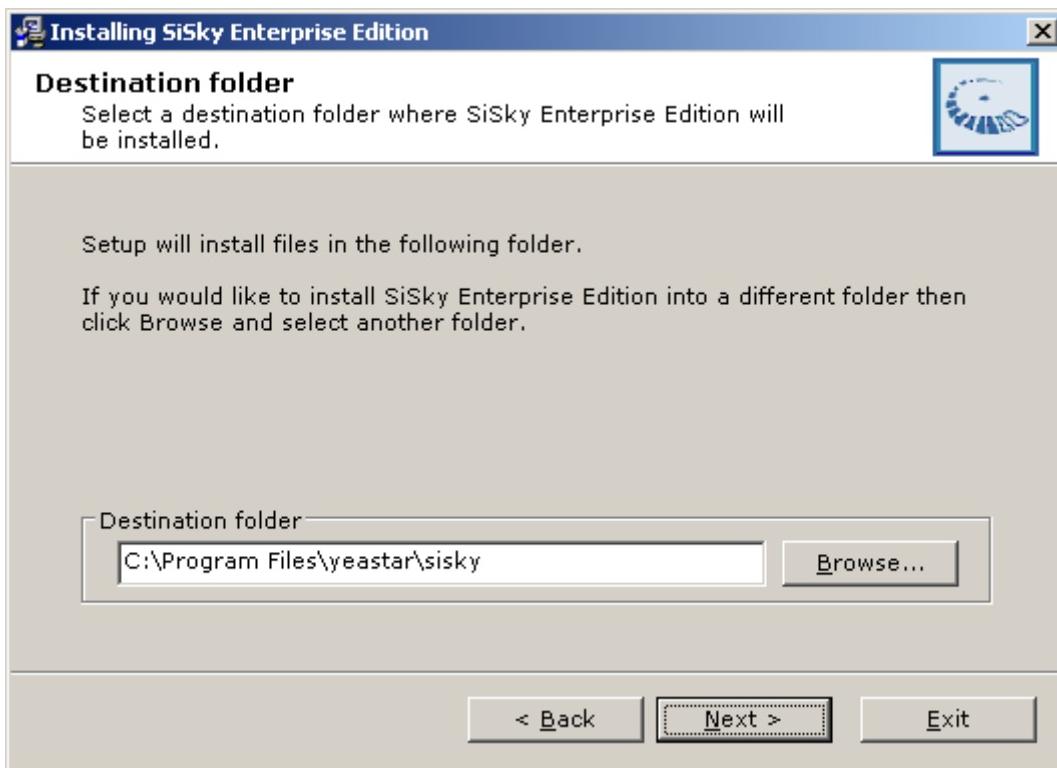


Figure 2-4

6. Enable the options by your own demands, and then click **Next**.

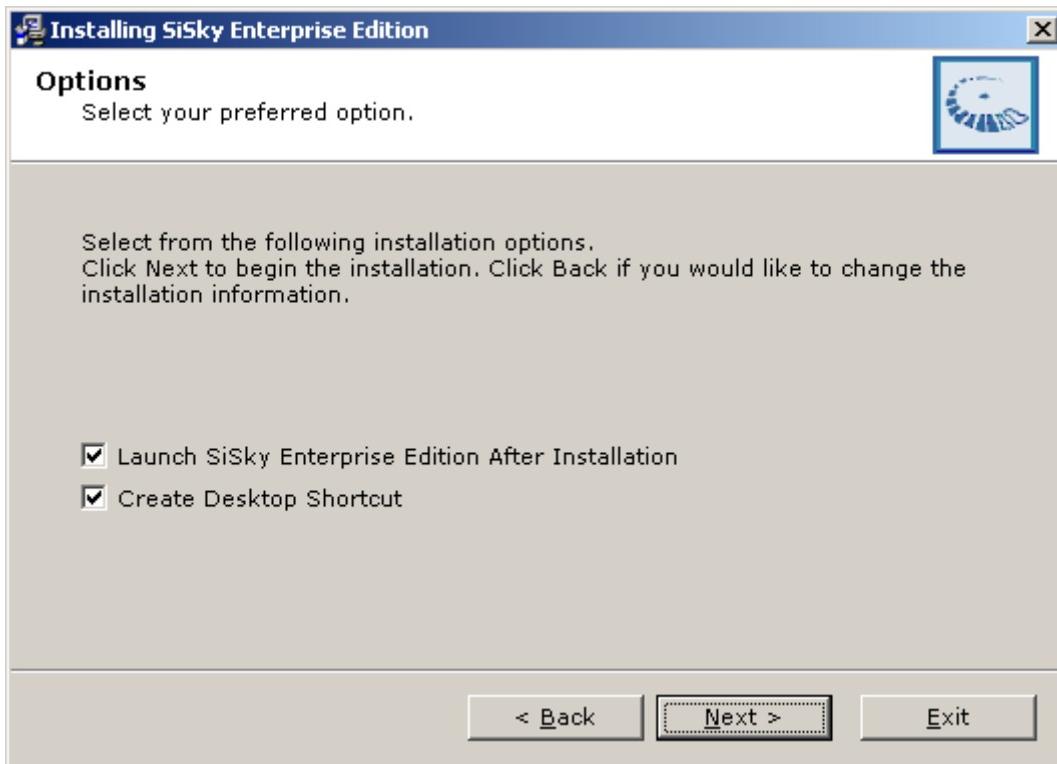


Figure 2-5

7. Enter into the **Installing Files**, system begin to configuration, which will last for a

while.

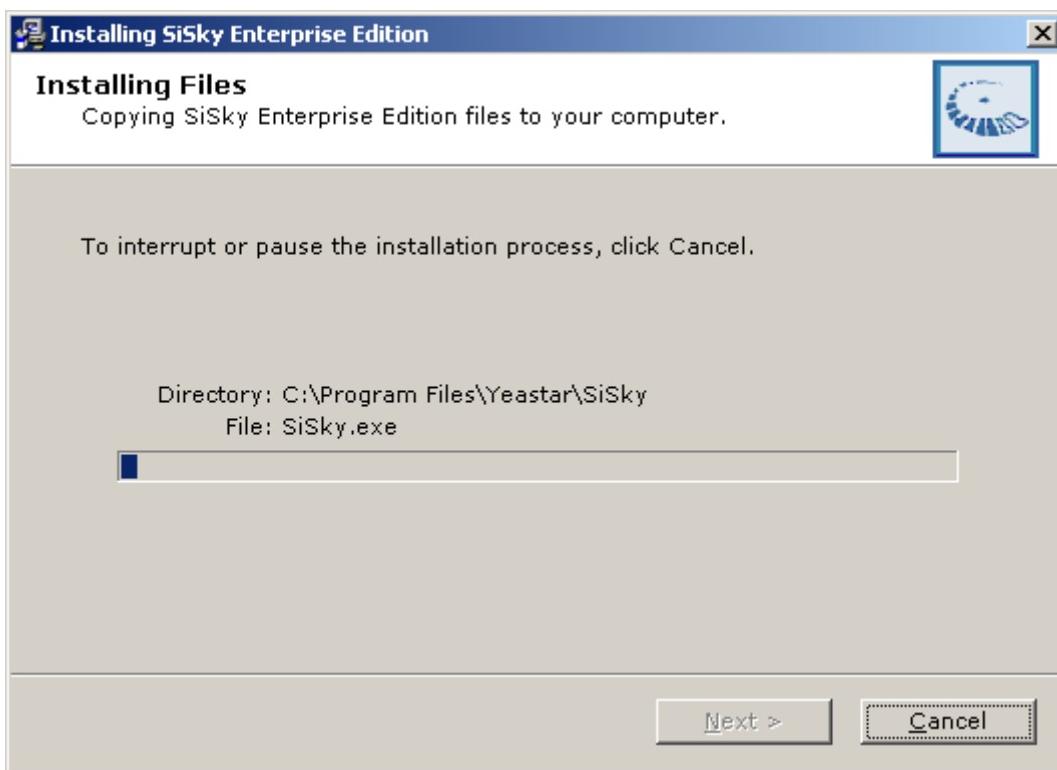


Figure 2-6

8. When the installation is complete, a screen will pop-up to notify you that the software is installed successfully. Click **Finish**.

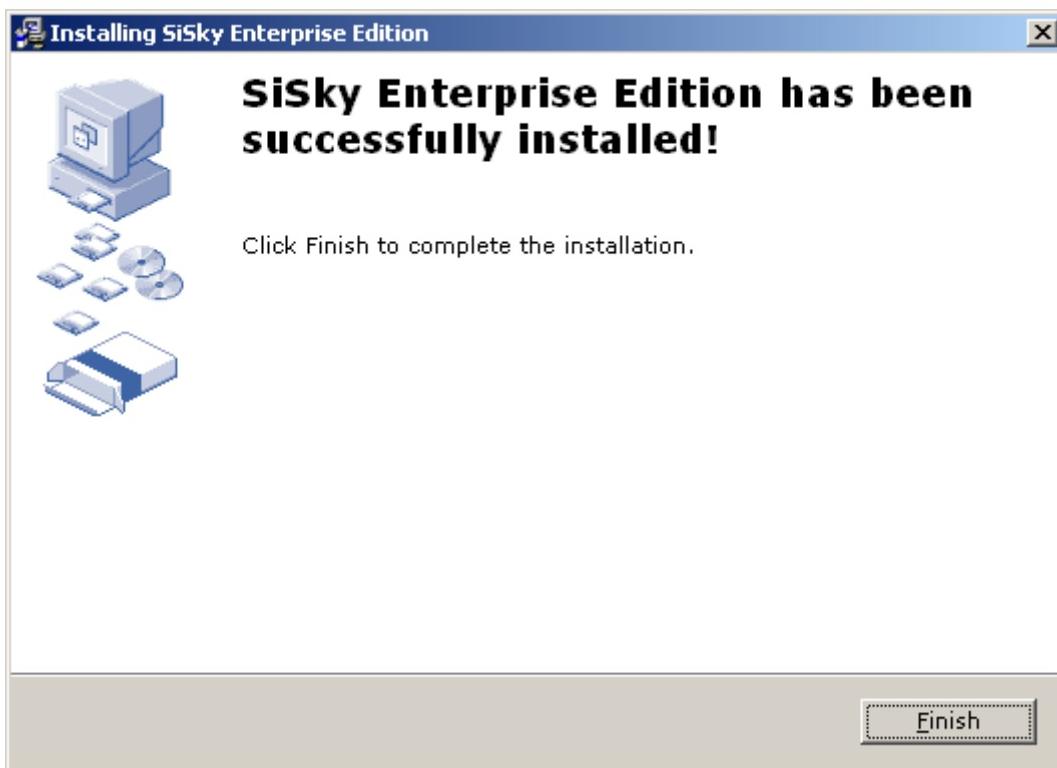


Figure 2-7

9. The final screen reminds you to restart computer in order to complete the installation.

You would better to restart now.



Figure 2-8

10. Launch SiSky to enter the next chapter.

## 2.2 Configuring Skype Accounts

### Step 1: Configure SiSky through Wizard

On initial use, a **Message** screen will pop-up and click 'Yes' to launch the Wizard.



Figure 2-9

Or you can click the **Config** on SiSky to launch the Wizard. (If the **Config** button is invalid, please click **Stop** to stop SiSky first)

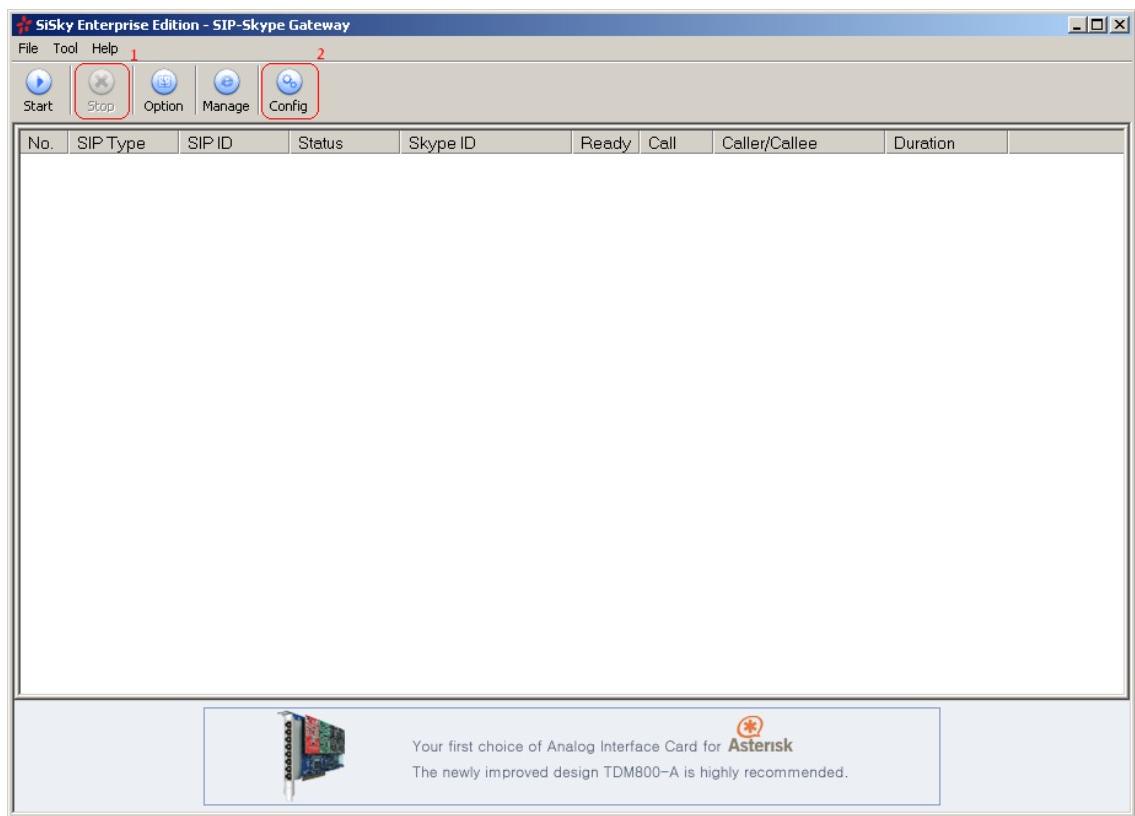


Figure 2-10

## Step 2: Launch the Config Wizard

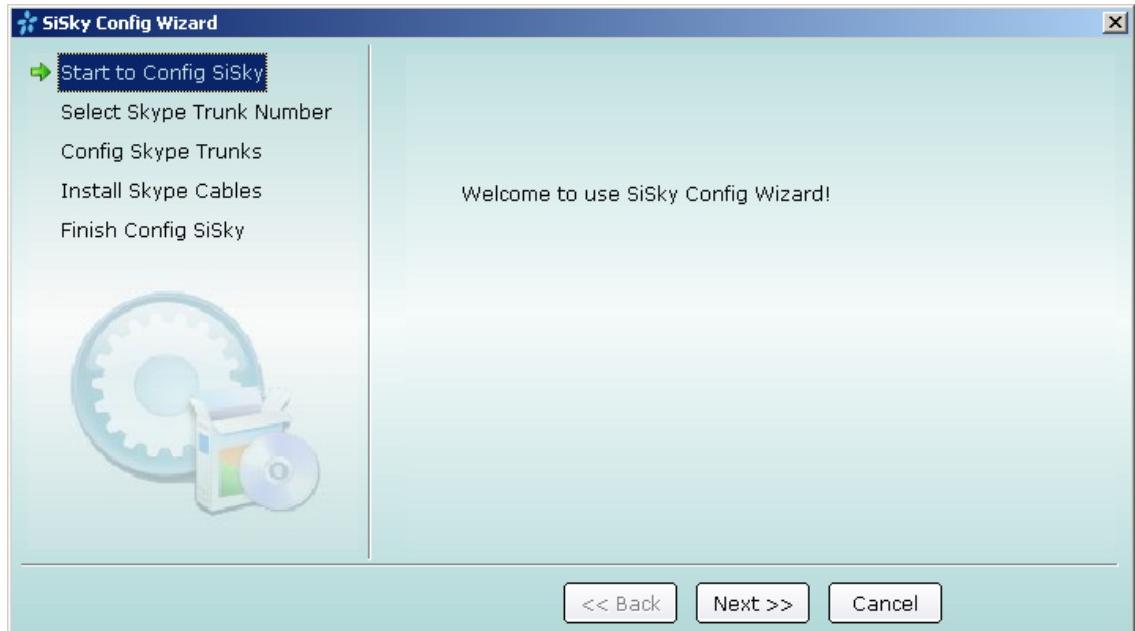


Figure 2-11

## Step 3: Select Skype Trunk Number

Select the **Skype Trunk Number** and then click **Next**

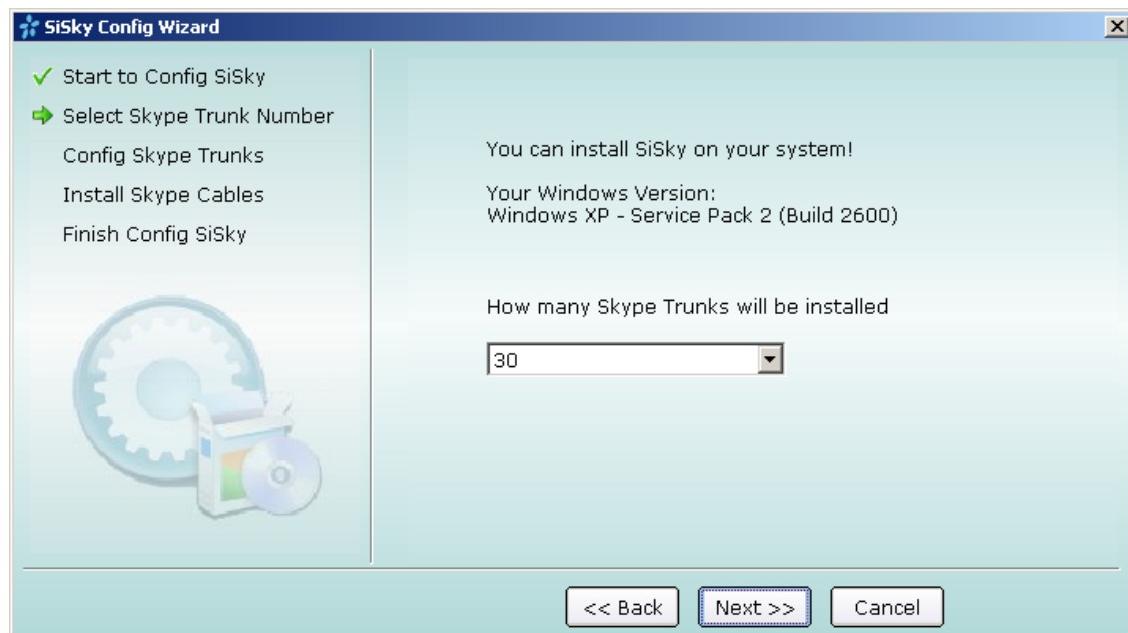


Figure 2-12

#### Step 4: Configure Skype for Each Port

Enable **Install Skype for this port** and Skype will launch.

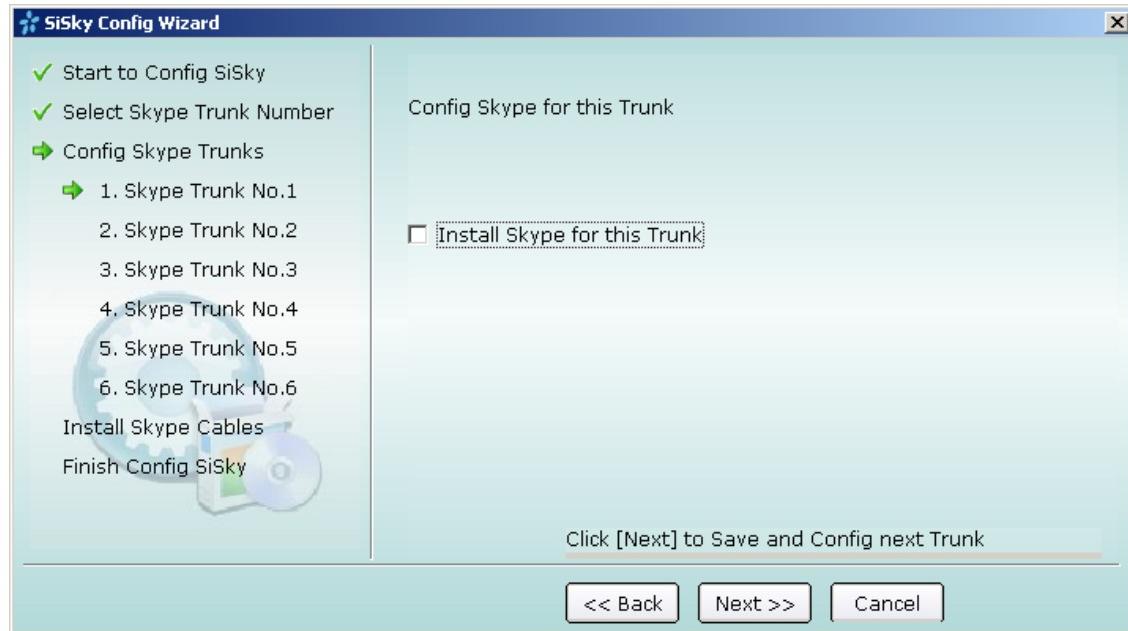


Figure 2-13

A **Skype?—Create Account** will appear. Create a new account (see Figure2-14) or Cancel it and log in by using an existing Skype account.

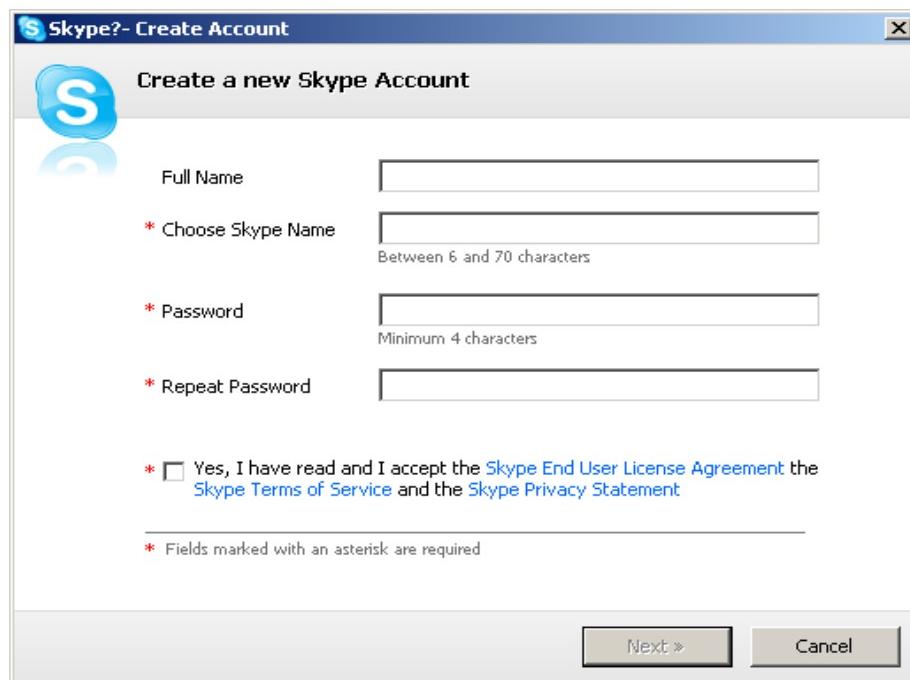


Figure 2-14

Enable the **Sign me in when Skype Starts** and wait for Skype to log you in.



Figure 2-15

When SiSky get the Skype Name automatically, the configuration of Port 1 is finished. Click **Next** to configure other ports by the same way.

**Note:** After these steps are complete, repeat step4 to configure the remaining ports and their Skype accounts. When the remaining ports are configured, there will be a green tick before every port. See Figure 2-16

## Step 5: Install Skype Cables

Enable **Install Skype Cables** and click **Next** to continue.

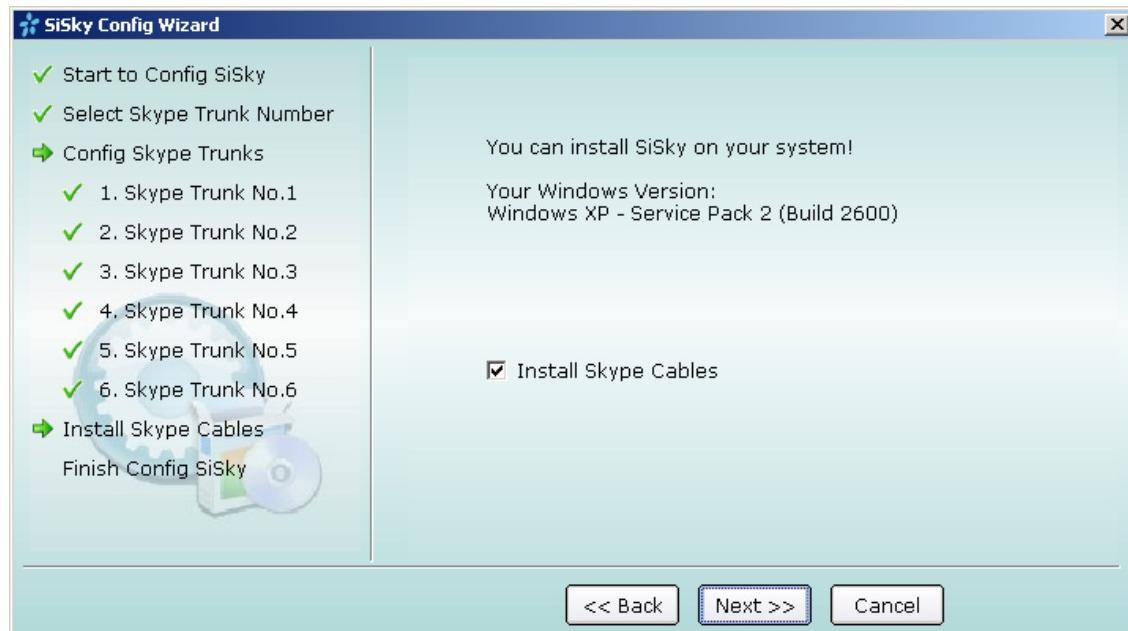


Figure 2-16

The following window maybe will appear during the system installation. Click **Continue Anyway**



Figure 2-17

## Step 6: Finish Config. Wizard

Select the country you are living, and then click 'Finish'.

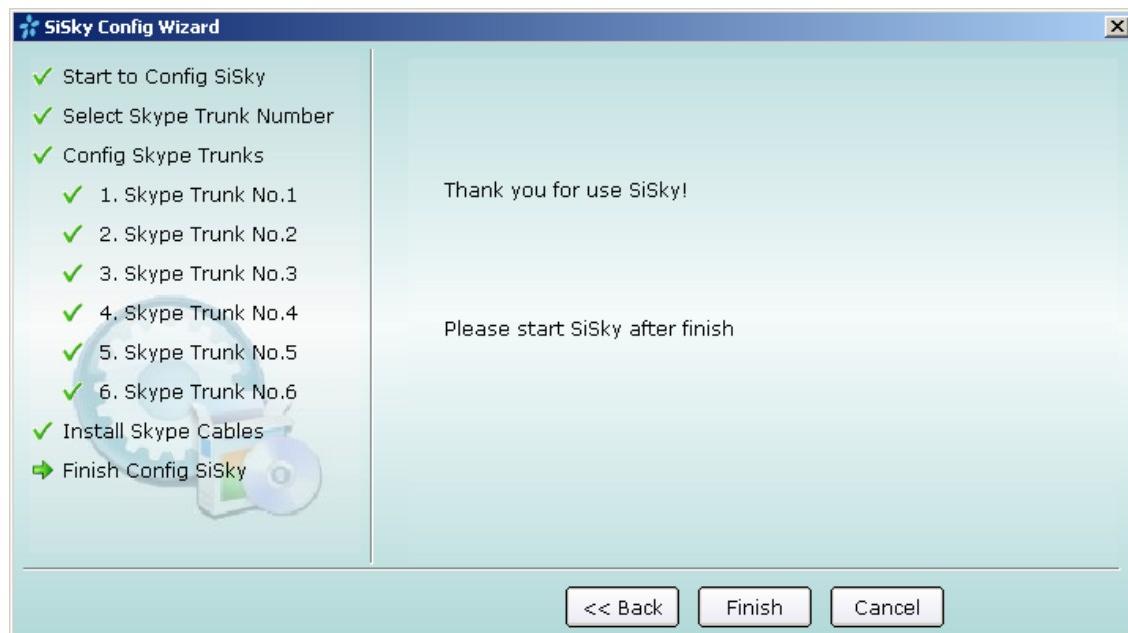


Figure 2-18

### Step 7: Double-click the shortcut on desk to run the SiSky software

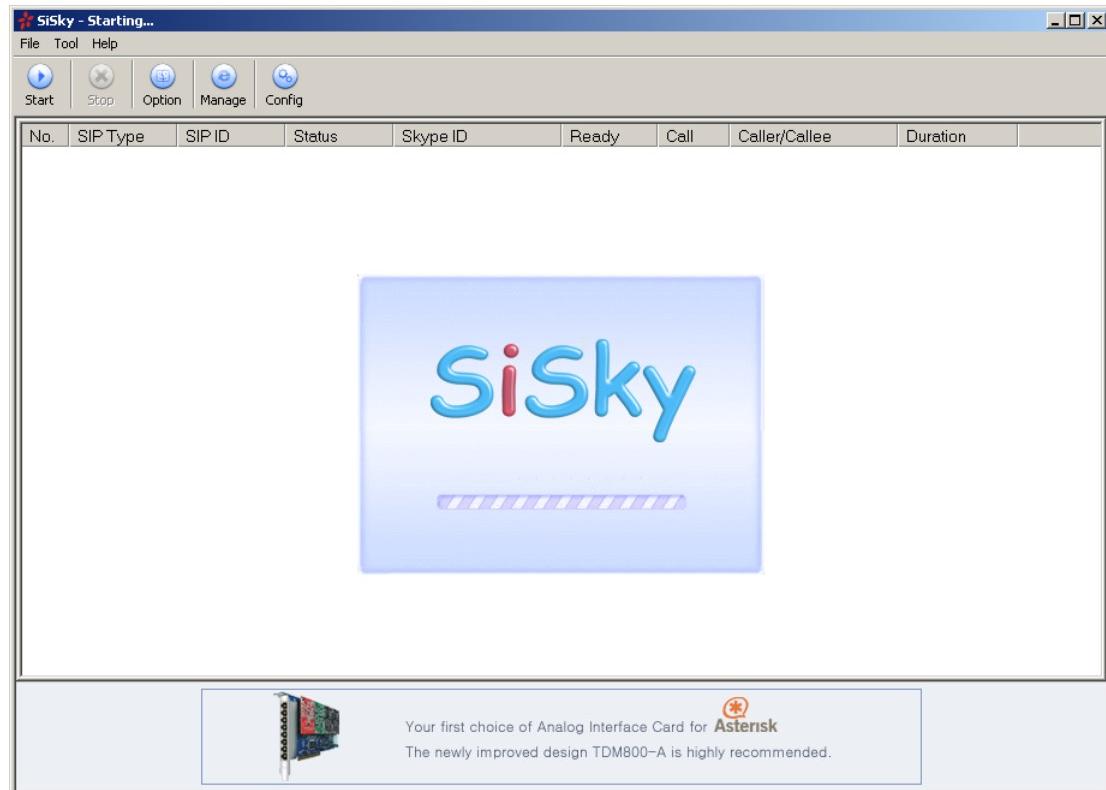


Figure 2-19

**Step 8:** Wait for all Skype IDs to log in. **Another program wants to use Skype** screen will come up. Click on the circle next to the first option, Allow this program to use Skype, and click **OK** to save.

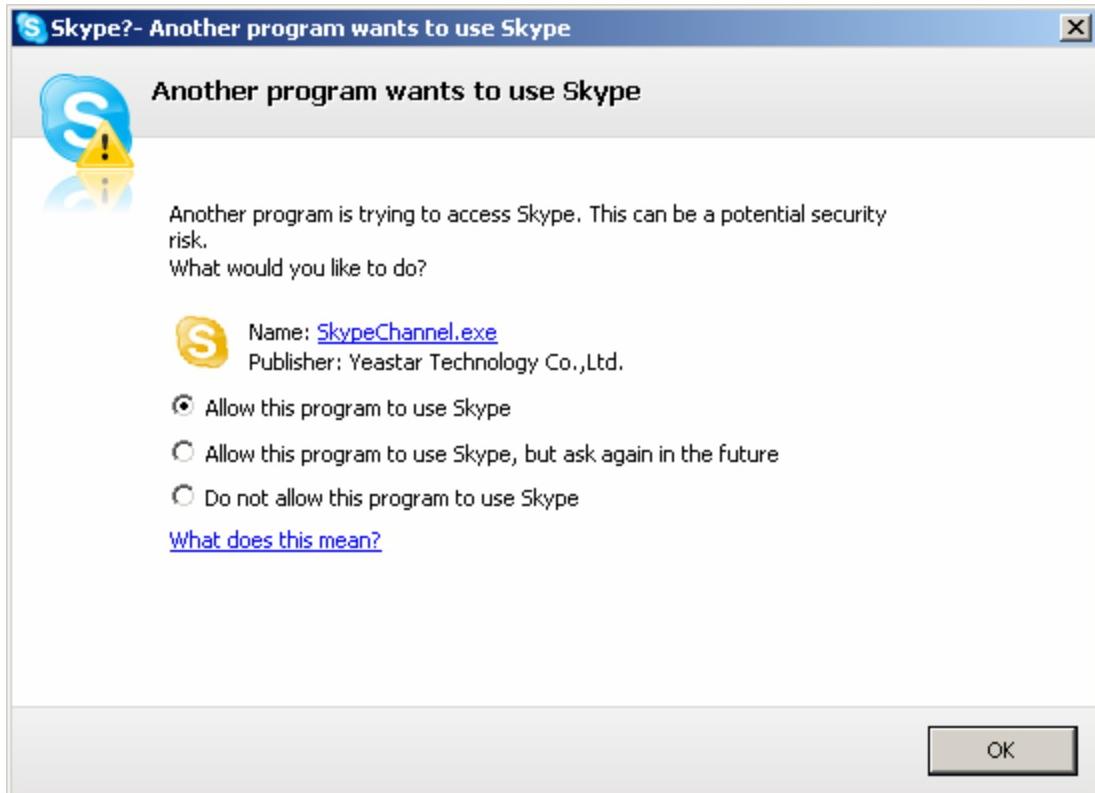


Figure 2-20

**Step 9:** Check if the status of all ports and Skype are absolutely normal as Figure 2-21. If Skype is launched successfully, the status of 'Ready' will show as 'Yes' on SiSky main interface.

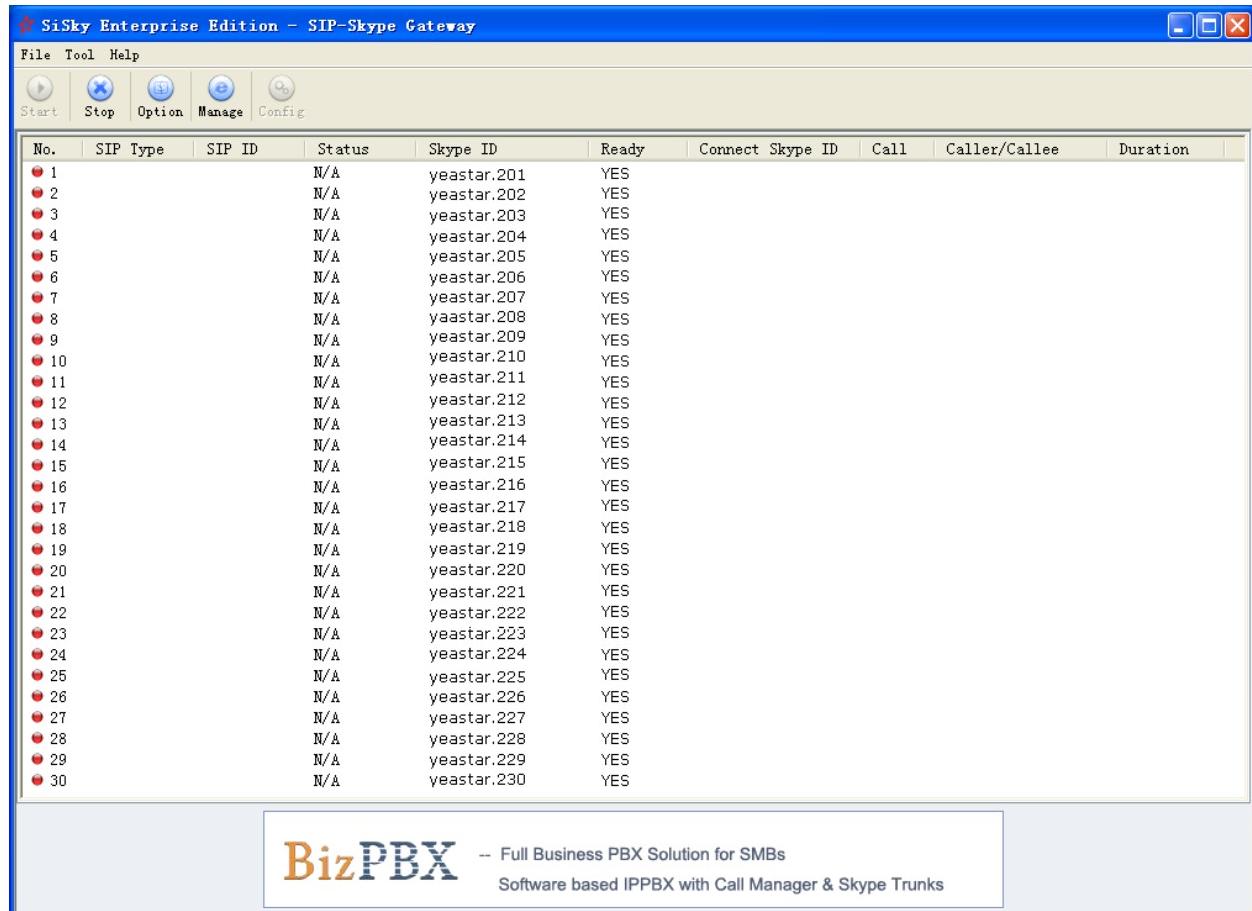


Figure 2-21

### 3. Installing VPS Soft Switch

#### Installation Steps:

##### 1. Download Software

Download VoipSwitch from official website <http://www.voipswitch.com>

##### 2. Installing VPS

Please refer to VoipSwitch User Manual for details.

## 4. Connecting Soft Switch with SiSky Server

For instance, VPS server's IP address is [192.168.5.240](http://192.168.5.240), SiSky server's IP address is [192.168.5.246](http://192.168.5.246). Take the voice code G711 as an example.

### 4.1 Configuring SiSky Server

#### 4.1.1 Set SiSky Options

##### 1. Set the Access IP Address

Click 'Option' on SiSky main menu.

1) Start SiSky When I start Windows

2) Add load balancing engine:

**Note:** this function will help system to equal call rate among all trunks.

3) Answer SIP incoming call immediately

**Note:** If disable this function, system will not sent reversal signal.

4) Access IP Address

**Note:** Multiple IP Address separated by ';'.

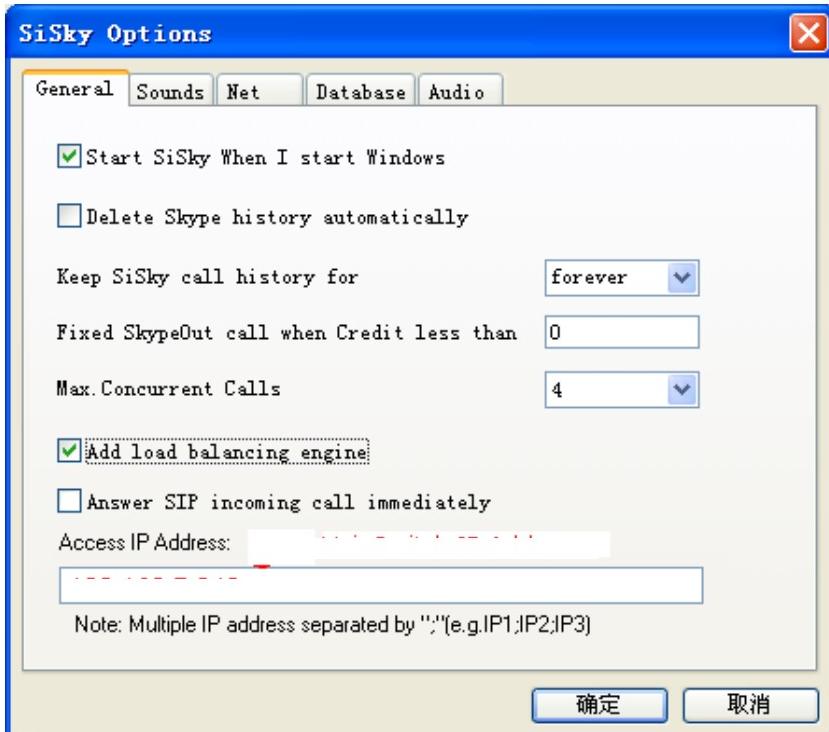


Figure 4-1

##### 2. Net

1) Enable Direct Outward Dialing-One of Extension Mode: P2P mode.

2) Use STUN Server: Enable STUN Server

Note: If server is under router, please enable this function.

### 3) Local Port: SIP port

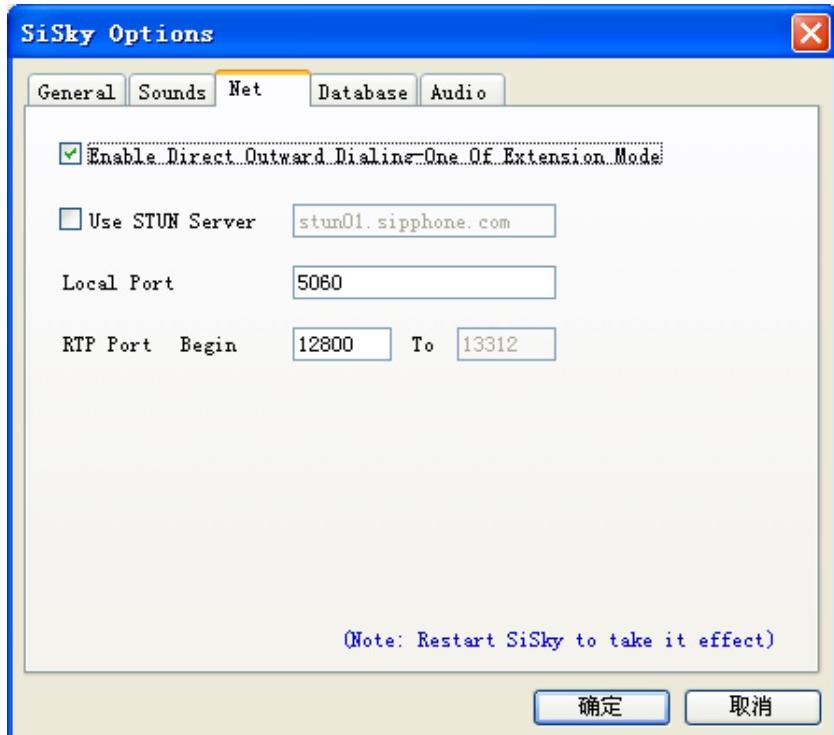


Figure 4-2

### 3. Audio

SiSky default code is G711. If you want to use G729 or G723, please click on the box and up it to the first rank.

Note: You need to restart SiSky software to activate the new configuration.

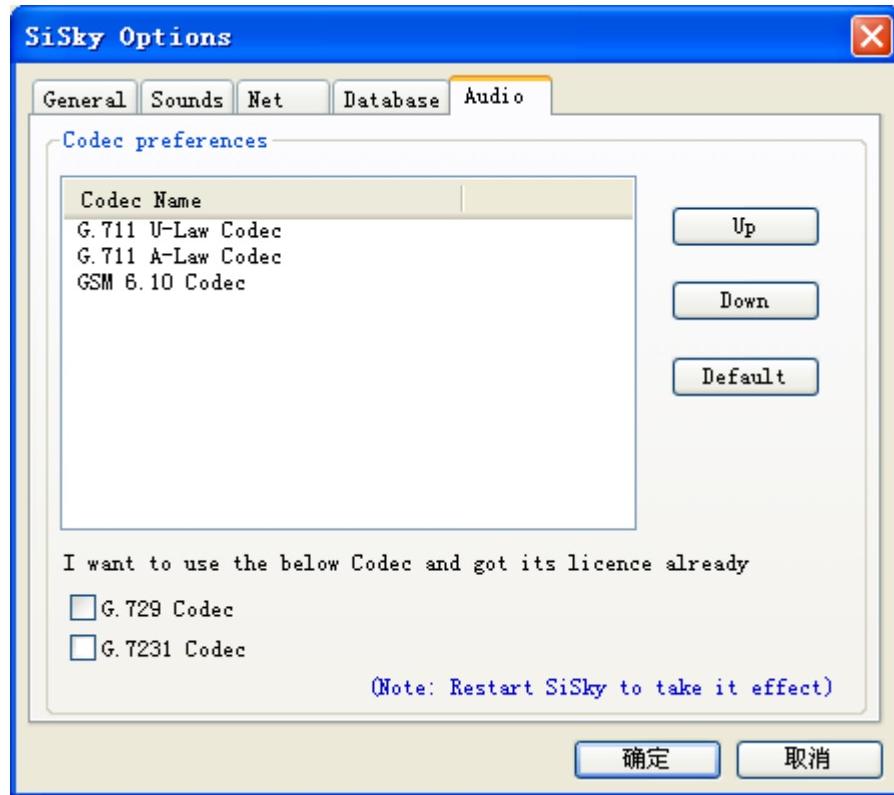


Figure 4-3

## 4.1.2 Set Dial Rule for SiSky

**Step1:** Open the SiSky Administration page.

Click 'Manage' button to enter into the page.

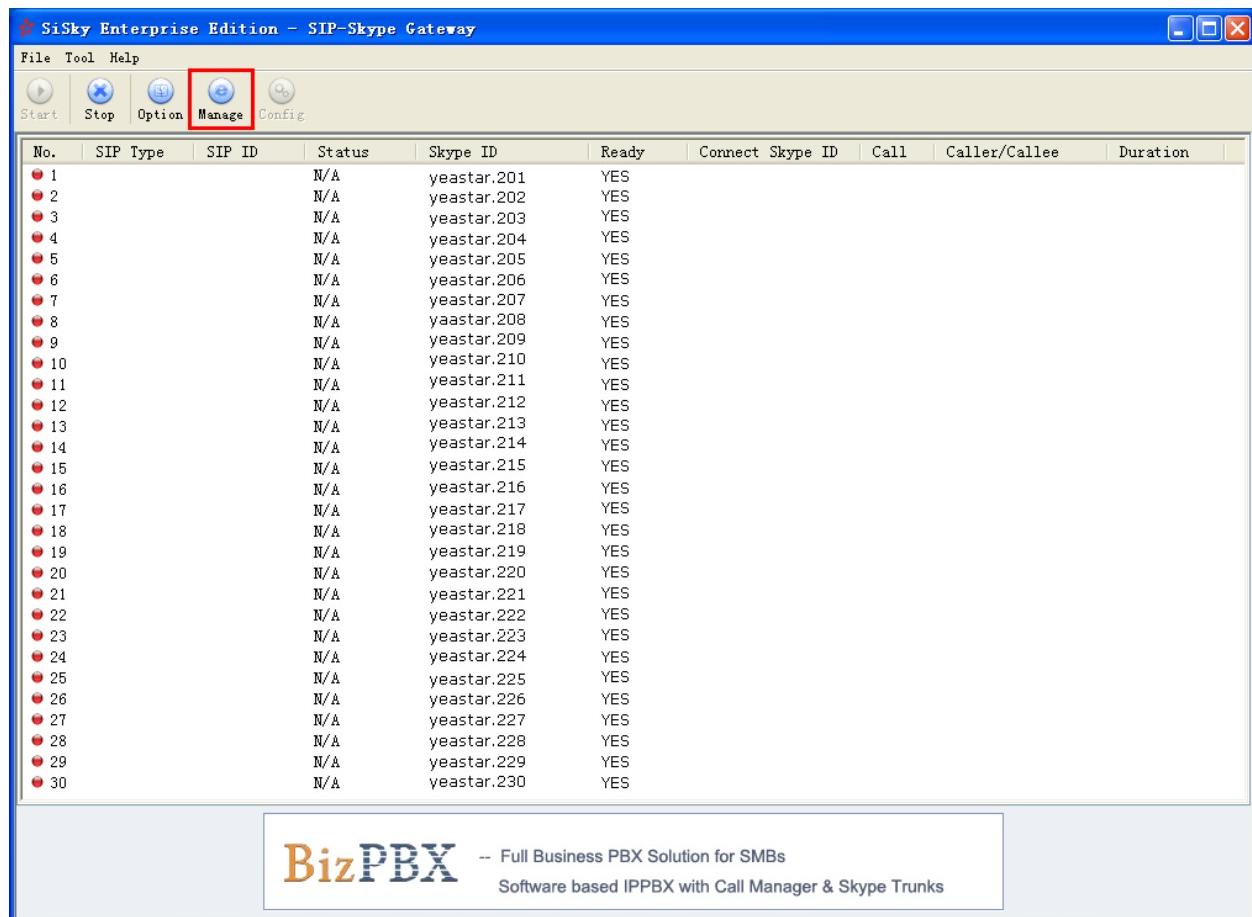


Figure 4-4

**Step 2:** Admin Log In

Enter the user name and password

The default user name is **admin**, and password is **password**

**Step 3:** Set Dial Rule

Making calls through Skype, one have to conform to the Skype dialing scheme as well as calling through PSTN. Maybe you had already got used to PSTN scheme and not accustomed to Skype rules. Therefore, **Dial Rule** settings will assist you to make Skype calls based on traditional PSTN calling habits. Keep the dialing habit as same as PSTN.

For example, a PSTN user in the mainland of China wants to make an international call IDDD (international direct distance dialing), which format is **00+country code+area code+telephone number**; make a domestic call DDD (domestic distance dialing), which format is **0+area code+telephone number**; While the SkypeOut format, no matter of

---

domestic or international calls, is **00+country code+area code+ telephone number.**

#### An International call:

1) A Chinese user calls to Canada in traditional **PSTN format**: the country code is 1, the area code is 416, phone number is "12345678", then he would dial: **00 +1+ 416 + 12345678;**

2) A Chinese user calls to Canada in **SkypeOut format**: the country code is 1, the area code is 416, phone number is "12345678", then he would dial: **00 +1+ 416 + 12345678;**

In this example, the PSTN and SkypeOur format for an international call is the same.

#### A Domestic Distance call:

3) A Chinese user in Paking calls to Shenzhen in traditional **PSTN format**: the area code is "755", phone number is "12345678", then he would dial:

**0+755+12345678;**

4) A Chinese user in Paking calls to Shenzhen in **SkypeOut format**: the area code is "755", phone number is "7571234", then he would dial:

**00+86 (country code)+755+12345678;**

Therefore, in order to not change the PSTN dialing format, this user can take advantage of dial rule to set it as below for simple:

substitute **00** for **00**; and substitute **0** for **0086** as Figure 4-5:

When the number begins by 00, SiSky will identify it as international call requirement. When the dialing call begins by 0, SiSky will identify the 0 as domestic call requirement and transfer it to 0086 automatically to conform SkypeOut format.

Why we substitute 00 for 00? Say it simply, to prevent 00 matches with 0. Because both the international and domestic calls are begin by 0, for fear of replacing **00141612345678** with **00860141612345678**.

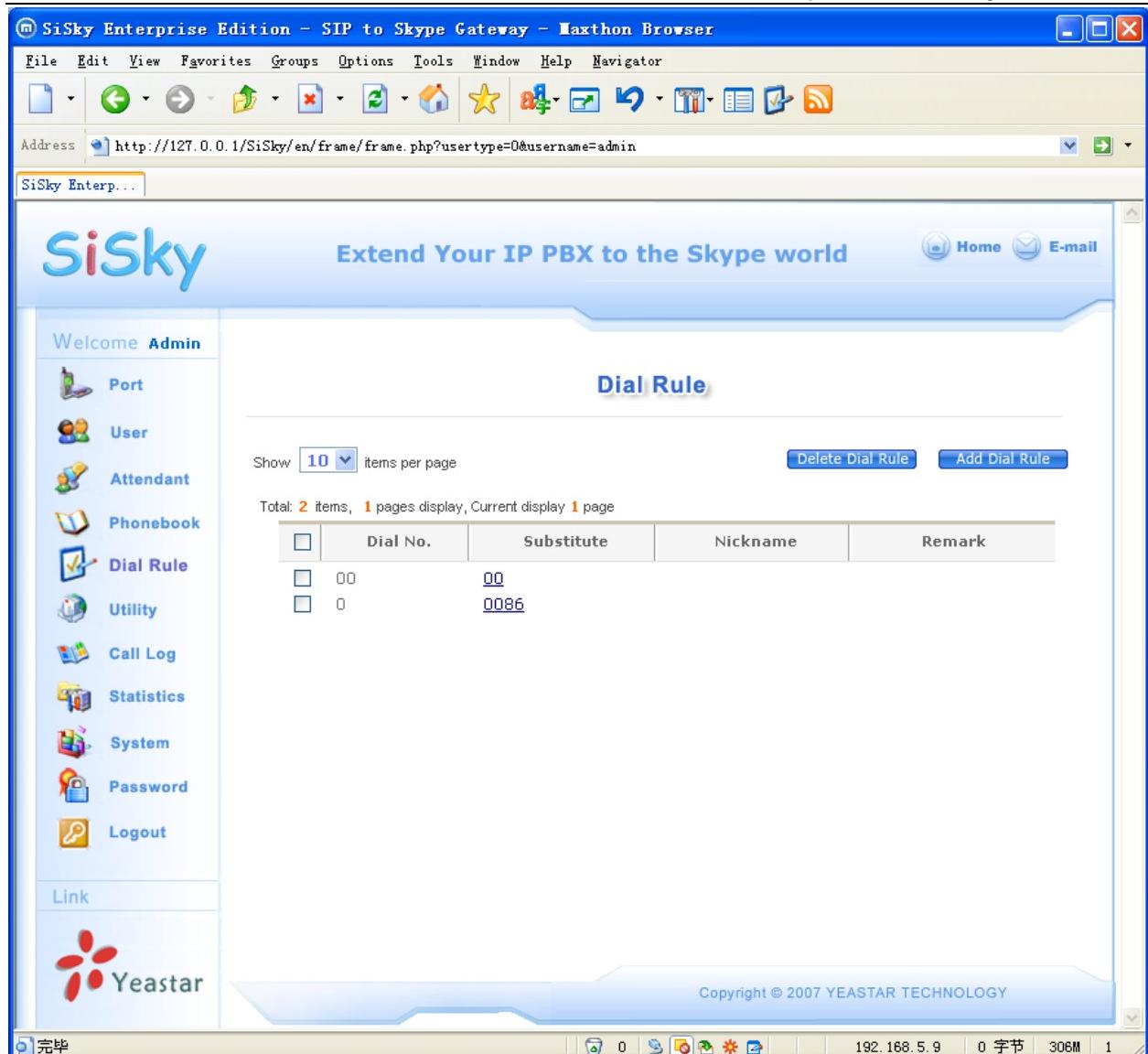


Figure 4-5

#### 4.1.3 Set Skype incoming call route in SiSky

**E.g.** Configuration the **Port1** incoming route.

**Step1:** Open the SiSky Administration page → Port configuration page.

**Step2:** Choose Port1.

Example, If have skype user make callin, let it call to SIP Server's Extension number or IVR Number **7777**, and the SIP Server's IP Address is 192.168.5.246. so the configuration like below.

**Direct In:** 7777@192.168.5.246 (extension number@sip server's ip address)

**Busy Transfer:** selected "To Any Idle Port in this server"

**Note:** other ports settings like port1.

## Port List

### Port 1 Setting

Trunk Error! Please ensure if it is available.

#### SIP mode

<input checked="" type="radio"/> <b>Work as Asterisk'/IPPBX's Extension</b> SIP Account: <input type="text"/> Password: <input type="password"/> SIP Proxy: <input type="text"/> : 5060 SIP Domain: <input type="text"/> DTMF Mode: <input type="button" value="RFC 2833"/> <input type="button" value="TTS"/> Expire Time: <input type="text" value="1800"/> STUN: <input type="button" value="Disable"/> <input type="button" value="Enable"/>	<input type="radio"/> <b>Work as Asterisk'/IPPBX's Trunk</b> SIP Account: <input type="text"/> Password: <input type="password"/> DTMF Mode: <input type="button" value="RFC 2833"/> <input type="button" value="TTS"/> STUN: <input type="button" value="Disable"/> <input type="button" value="Enable"/>
---	--

#### Skype Profile

Allow this Skype Status to be shown to everyone

Use Skype: Yes

Skype Status: No

Skype ID:

Skype Balance: 0

#### Other

Extension Number@SIP Server IP Address

Direct In:

(always dial this number for incoming calls)

Direct Out:

(always dial this number for outgoing calls)

No Transfer

Busy Transfer:  To Any Idle Port in this Server

To Customized Skype ID

Figure 4-6

## 4.2 Set VPS

This section only introduces the connectivity part of VPS and SiSky. If you have any other questions regarding the configuration of VPS, please refer to VPS User Manual.

### 4.2.1 Connect VPS to SiSky Server

Please choose 'other' → 'Gateways' on VPS left listing menu. Create a new Gateway to dock in SiSky server.

#### 1. Gateway data

Description: SIP-Skype (the definite name of Gateway)

IP Number: 192.168.5.246 (SiSky server's IP)

Port: 5060 (SiSky's UDP port)

Calls limit: 10 (decide by user own demands)

Active: Yes (Whether or not to use this Gateway)

#### 2. Connection properties

Supported codecs: G711 (here we just take G711 as example)

Please empty SIP Device, Username and Password items

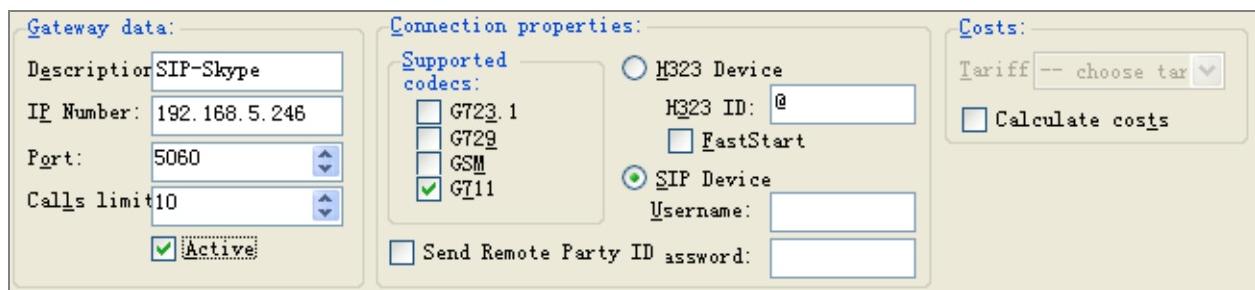


Figure 4-5

### 4.2.2 Set VPS Dial Rule

Choose 'Dialing plan' on the VPS left listing menu, and create new dial rule:

Steps are as following:

1. Number: 0
2. Priority: 0
3. Destination: External gateway, and choose 'SIP-Skype'.
4. Proxy Settings:  
SIP Client → Media proxy

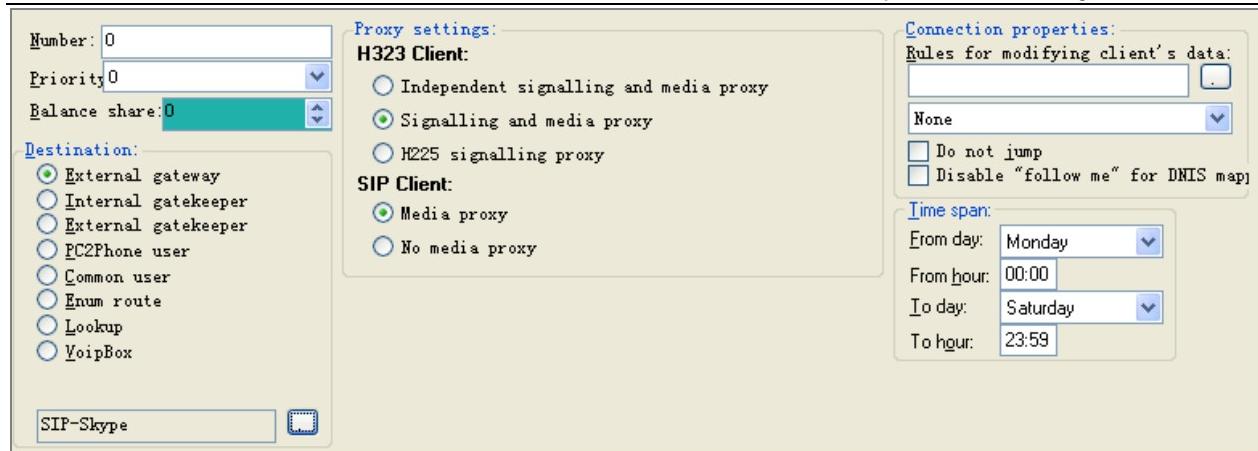


Figure 4-6

#### 4.2.3 Set Tariffs

- Choose 'Tariffs' on VPS left listing menu, and create a new Tariff and name it as SkypeTariff.

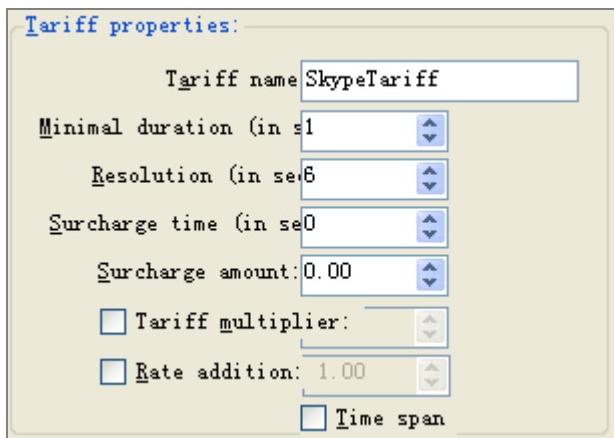


Figure 4-7

- Edit SkypeTariff and create a new prefix that begins by '0'.

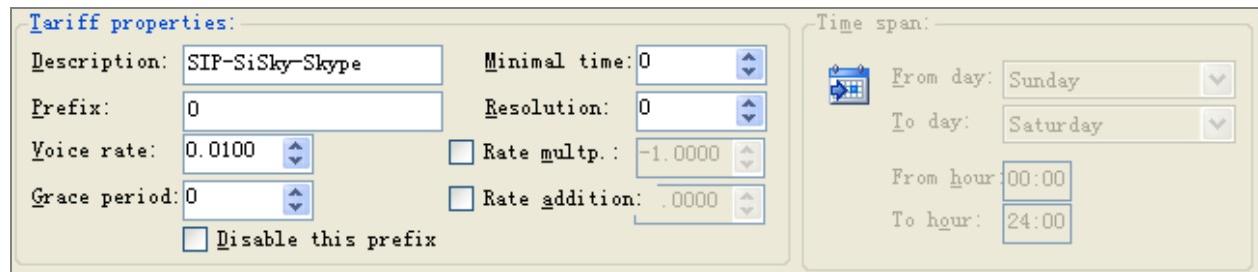


Figure 4-8

#### 4.2.4 Set GK/Registrar Clients

Choose 'Clients' → 'GK/Registrar Clients' on VPS left listing menu and create 'GK/Registrar Clients'.

1. Take an example, user create an new account 'iphone'

Login: iphone

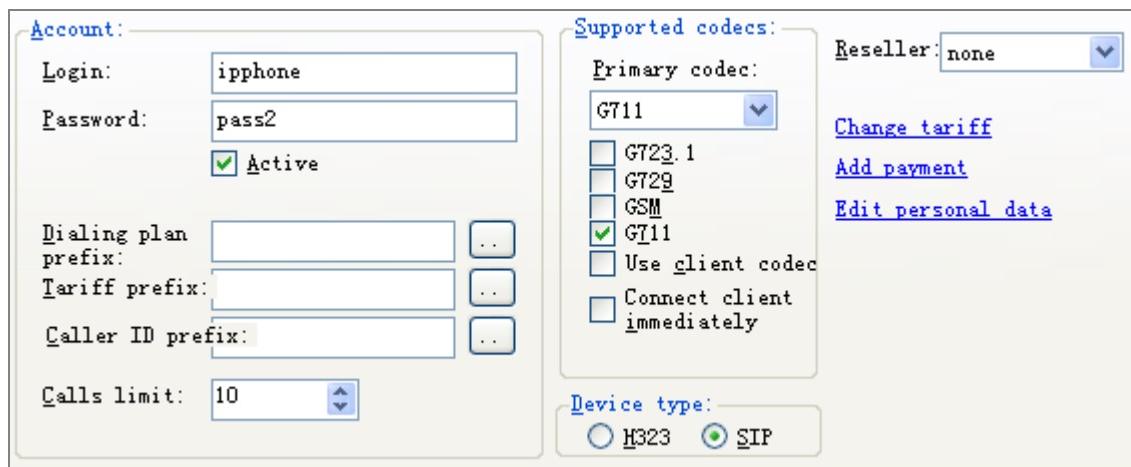
Password: pass2

Active: yes

Calls limite: 10

Primary codec: g711

Device type: SIP

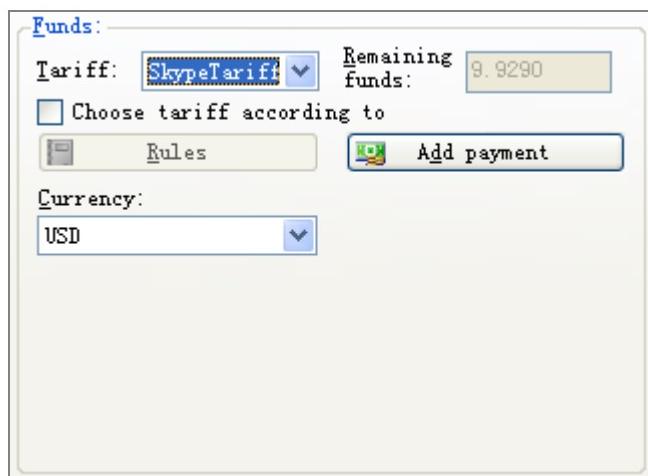


The screenshot shows the 'Account' configuration for a client named 'iphone'. The 'Primary codec' is set to 'G711'. Other codecs listed include G723.1, G729, GSM, and G711. There are also checkboxes for 'Use client codec', 'Connect client immediately', and 'Device type' (set to 'SIP').

Account:		Supported codecs:	Reseller:
Login:	iphone	Primary codec:	none
Password:	pass2	G711	<a href="#">Change tariff</a>
<input checked="" type="checkbox"/> Active		G723.1	<a href="#">Add payment</a>
Dialing plan prefix:		G729	<a href="#">Edit personal data</a>
Tariff prefix:		GSM	
Caller ID prefix:		G711	
Calls limit:	10	Use client codec	
		Connect client immediately	
		Device type:	
		H323	SIP

Figure 4-9

2. Assign tariff for extension. Here we assign 'Skypetariff' for this extension.



The screenshot shows the 'Funds' configuration. The 'Tariff' dropdown is set to 'SkypeTariff', and the 'Remaining funds' display shows '9.9290'. There is a checkbox for 'Choose tariff according to Rules' and a 'Add payment' button. The 'Currency' dropdown is set to 'USD'.

Funds:	
Tariff:	SkypeTariff
Remaining funds: 9.9290	
<input type="checkbox"/> Choose tariff according to Rules	
<a href="#">Add payment</a>	
Currency:	
USD	

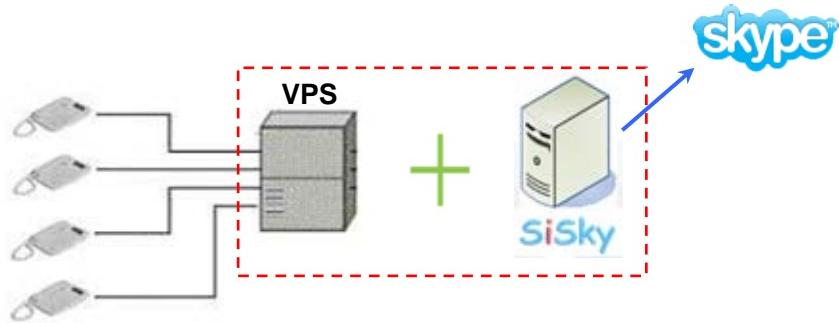
Figure 4-10

## 5. Sample

### 1. Application Demands

- 1) Utilizing SiSky as Skype landline for VPS
- 2) Make both domestic long distance and international calls at low rate

### 2. Application Environment



### 3. Test

#### 1) Make call out via Skype

If extension 'ipphone' wants to make calls through Skype, dial like this: 00 + country code + city code + telephone number or 00 + country code + mobile phone number.

If you want to dial like this: 0+city code + phone number or 0 + mobile phone number, please configure the Dial Rule in sisky, see chapter 3.1.2

#### 2) Receive the incoming call from Skype

Skype user make call to the SiSky server's skypeid, it will auto call to the extension number 7777.

[End]